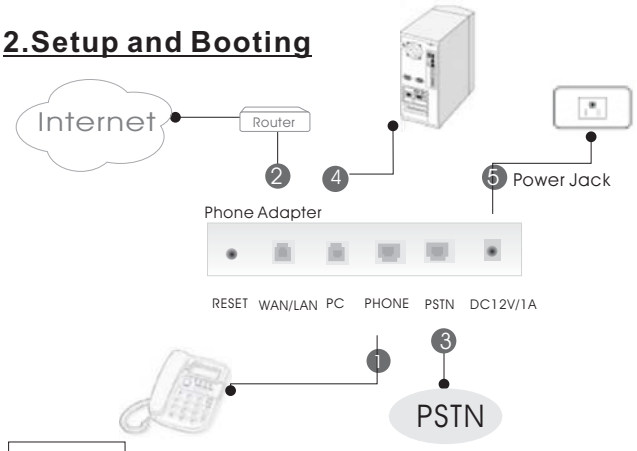
User Manual
VoIP Adapter GLVP-8000

Package Contents & Setup and for Booting

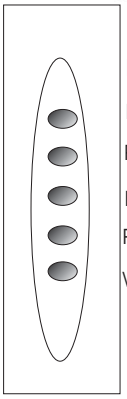
1.Package Contents

A. One Telephone Adapter
B. One 12V/500mA Power Adapter
C. One LAN Cord
D .One Telephone Line Cord

2.Setup and Booting



1.Plug telephone on PHONE socket.
2.Plug RJ45 line on WAN socket to connect to Internet.
3.Plug POTS line (if available) on PSTN socket.
4.Plug RJ45 line on LAN socket to connect to PC.
5.Plug power adaptor.
6.Wait for booting completed. A ring from the telephone indicates device booting successfully and begin registering to server.
7.When registering function OK. The LAN LED indicates in yellow.



LED's Indication:

Power on indication (green)
PSTN connection status (green)
Booting finish indication (green)Register OKindication(yellow)
PC port 10/100M link (green/yellow)
WAN/LAN port 10/100M link (green/yellow)

1.To make the phone call through firewall, you need to open SIP port and RTP port on your firewall. (Please also check the firewall setting of anti-virus software.)
2.Please check the Network Type (DHCP or PPPoE or Static IP) and finish the configuration process.
3.Press Reset Button arround 4 secs. to the default value if needed.

VoIP & PSTN Operation

VoIP Operation

Dial Peer IP Phone directly OR Dial E.164 through SIP Server

Step 1: Pick up phone and you will hear the dial tone.

Step 2: Dial IP address use '*' instead of '.' | Dial E.164 number

Step 3: press '#' to send

If IP address = 192.168.1.163
real dial key: 192*168*1*163#

If the peer phone's E.164 number is 100 real dial key: 100#

PSTN Operation

Make a PSTN call

Step 1: Pick up phone and you will hear the dial tone.
Step 2: Press the " * " key to switch to PSTN mode and you will hear the dial tone from PSTN
Step 3: Dial the number you want to make the call.
Step 4: Hang up the phone when the call completes and repeat step 1 to make another call.
Step 5: Hang up the phone for two seconds to switch Back to VoIP mode

VoIP phone calls the regular phone:

The GLVP-8000 can call any land or mobile phone anywhere in the world pls.
Dial with following description:
Step 1: Going into the Web configuration with correct password.
Step 2: Set the country code as you belong then you can dial the number as your country rule.
Step 3: Buy the credit from the VoIP of www.glinksnet.com before you make the call.
Notice: Pls. don't make the call to regular phone from the VoIP phone unless you had buy the credit from the G-Links' dialing out:

1.Go to the Web of the www.glinksnet.com
2.Click VoIP to go into the dialing out page(CCNET VoIP billing system).
3.Choice the language you want.
4.Enter your user name(the phone number) and pass code.
5.Begin to enjoy the G-Links' CCNET VoIP platform that you can buy the credit of dialing out, on the line and many other functions.

IVR Command Table

Char	Char Mapping
0 . @ : __ + -	00 01 02 03 04 05 06
1 Space "	10 11 12
2 abc ABC	20 21 22 23 24 25 26
3 def DEF	30 31 32 33 34 35 36
4 ghi GHI	40 41 42 43 44 45 46
5 jkl JK L	50 51 52 53 54 55 56
6 mno MNO	60 61 62 63 64 65 66
7 pqrs PQRS	70 71 72 73 74 75 76 77 78
8 tuvTU V	80 81 82 83 84 85 86
9 wxyz WXYZ	90 91 92 93 94 95 96 97 98

IVR Char Mapping Table
Configure IVR item with type IVR_CMD_TYPE_SET_STRING
Examples:
Example 1, user name is abc.
Input : 212223#
Example 2, user name is 123@isp.com
Input : 1020300243747101236361#
Playing IVR examples:
Example 1, " aBcDef "
Playing : " a up b c up d e f "

Call Hold:
Step1: Flash hook (key) and press "1"
Step2: Then you will hear dial tone.
Press the number you want to dial.
Step3: Flash hook (key) to switch
Between the two calls.

Call Transfer:
Step1: Flash hook (key) and press "2"
Step2: Then you will hear dial tone.
Press the number you want to dial.

Call Waiting:
Step1: When hearing a call waiting tone. Press
" Flash hook (key)+1" to pick up the
Coming call.
Step2: Press" Flash hook (key) " to switch between
the two calls.

Web Setting (To configure the device with browser)

The access buttons, located across the top of the web page, are used to configure the IP Phone for operation.

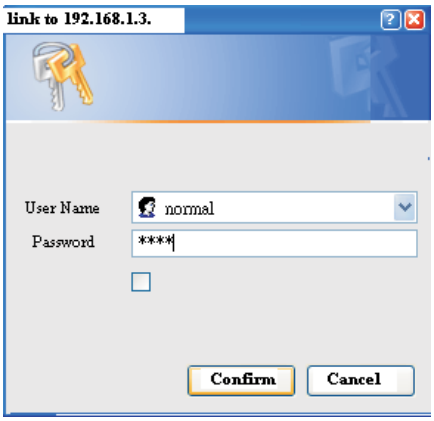
For example: 192.168.1.3

1.Please enter IVR Menu "16881688#"

-->"100#" to get your IP address

2.Please open MSIE(Win98,Win2000 and XP)
And input your IP address--http://xxx.xxx.xxx.xxx:8080

3. Input user name and password
user name is : **normal**
password is : **1234**



SIP Settings

Network Type:	Static IP
IP Address (Static IP):	220.130.187.221
Subnet Mask (Static IP):	255.255.255.0
Gateway (Static IP):	220.130.187.254
PPPoE User Name:	123
PPPoE Password:	●●●
NTP Server IP:	
NTP Time Zone:	(GMT+08:00)
NAT_TRAVERSAL	None
STUN URL:	stun.fwdnet.net
NAT external ip:	200.200.200.200
First DNS IP:	168.95.1.1
Second DNS IP:	168.95.1.1
Band Width Control (kbps):	1000

1. Network Type
Specify the Network type to connect to Internet
2. IP Address
Assign the IP address of WAN interface Effective only when network type is static IP; for example: 220.130.187.221
3. Subnet Mask
Assign the subnet mask of WAN interface Effective only when network type is static IP; For example:255.255.255.0
4. Gateway
Assign the default route.Effective only when network type is static IP; For example 220.130.187.254
5. PPPoE User Name/ Password
Effective only when network type is PPPoE For example: 123
6. NTP Server IP
Network Time Server IP. Used to get time Form this server
7. NTP Time Zone
Global Time Zone of your current location
8. NAT Traversal(none stun static)
If "None" is selected,no stun operation in SIP activities
If "Stun"is selected,the device will detect the NAT type automatically and get the result to modify SIP activities
Be aware to set the"Stun URL"when "Stun"is selected.
If "Static" is selected, specify the "NAT" external IP
9. STUN URL
STUN server domain name or IP address. When NAT TRAVERSAL is "Stun", you must set this item. For example, "stun.fwdnet.net" or "80.67.17.102".
10. NAT External IP
IP address of the NAT device in front of the Device
11. First DNS IP
Primary Domain in Name Server used to lookup IP address form domain name
For example:"168.95.1.1"
12. Second DNS IP
Second Domain in Name Server used to lookup IP address form domain name
For example:"168.95.1.2"
13. Band Width Control: used to control the traffic rate flooding between LAN/WAN interface, in order to reserve bandwidth for VoIP. For example,1000kbps, It means that the max traffic rate will be limited to 1000kbps in Bridge/NAT mode

SIP Settings

SIP Settings are needed under the phone bundled with a number(from service provider)

Using Proxy Server:	(•) On ( ) Off
Proxy url:	210.202.244.130:7890
Proxy user:	98600004
Proxy Password:	••••••
User Name:	98600004
Using Outbound Proxy Server:	( ) On (•) Off
Outbound Proxy url:	
Domain Name:	210.202.244.130:7890
Expire Time:	3600
RTP PORT:	0

1. Using Proxy Server
   Choose to use proxy server or not

2. Proxy URL
   If "Using Proxy Server" is selected this item must be identify
   For example:
   "210.202.244.130:7890"
   When the SIP server does not use the default port number, you can specify the port number by the format "IP:Port Number"
   For example,
   "210.202.244.130:7890"

3. Proxy user
   The SIP account name when"Using Proxy Server" is on.

4. Proxy password
   The SIP account password when "Using Proxy Server" is on

5. User Name
   User name of SIP user agent

6. Using Outbound Proxy Server
   Specify to enable or disable outbound Proxy server

7. Outbound Proxy URL
   Effective only when "Using outbound Proxy server" is on

8. Domain Name
   Used when "Using Proxy Server"is on

9. Expire Time
   The period of registering time in second

10. RTP Port
    Specify the RTP port number. If "0" or bland is set, the system will choose one Port at random

NAT Relative

NAT Enable:	(•) On ( ) Off
LAN IP Address:	192.168.1.3
LAN Subnet Mask:	255.255.255.0
DHCP Server Enable:	On
DHCP Server Start IP:	192.168.1.200
DHCP Server IP Count:	50

Virtual Server 1 :	Port		Protocol	TCP	Local IP
Virtual Server 2:	Port		Protocol	TCP	Local IP
Virtual Server 3:	Port		Protocol	TCP	Local IP
Virtual Server 4:	Port		Protocol	TCP	Local IP
Virtual Server 5:	Port		Protocol	TCP	Local IP

1.NAT Enable
Select "On" to enable NAT function;
Select "Off" to disable NAT function and Enable bridge function;
When "NAT Enable" the DHCP server function is disabled even that Configured it at "On" .

2.LAN IP Address
Default Value:192.168.111.1

3.LAN Subnet Mask
Default Value:255.255.255.0

4.DHCP Server Enable
Select "On" to enable DHCP server on LAN Interface. Select " Off" to disable this function.DHCP server will automatically shut down when NAT is disabled .

5.DHCP Server Start IP
Default value:192.168.111.100

6.DHCP Server IP Count
Default value: 30

7.Virtual Server 1~ 5
Virtual servers setting for NAT.
For example, one FTP server behind NAT router with IP address:
"192.168.111.100"
Port:21
Protocol:TCP
Local IP:192.168.111.100

Phone Features

Display Name:	user
Unconditional Call Forward:	
Busy Call Forward:	
No Answer Call Forward:	
No Answer Ring Time:	60
Using Call Waiting:	( ) On (•) Off

1.Display Name
The display name setting on SIP user agent

2.Unconditional Call Forward
Setting this item to forward every incoming call to the specified URL.
If the item is blank, incoming calls will not be forwarded

3.Busy Call Forward
Setting the item to forward the calls ,then the phone is busy
If the item is bland, the incoming calls when You are busy will be rejected

4.No Answer Call Forward
Setting the item to forward the call when no one pick up the calls in some times.
The item "No Answer Ring Time"is used to specify the time in second

5.No Answer Ring Time
Reference 4.4

6.Using Call Waiting
Select to turn on or off this feature
When the item is turned on, the incoming call will signal the alerting tone to you.

FPA (Frequent problems and Answer)

Problem	Possible Cause	Solution
Low Voice Quality	Your Network connection Bandwidths is less than 10Kbps Unstable Network connection : Network delay > 500ms or Trembling > 100ms or Network packet loss > 10%	Pls. check your LAN cord plug to make sure your plug in good condition and also check the Router or Switch Hub which you connect to, and make sure that you use the phone in the best condition network possible.
Can not log in to SIP Server (message : Registering Fail, no @ symbol in the position 'b' of LCD)	Log in obstructed by Firewall. You are using a proxy network which is blocking the log in. Your are not in DHCP network. You did not enter the correct configuration.	Pls. contact the G-Links customer center (technical@glinksnet.com / tomth@glinksnet.com). Pls. check the configuration which you set again. Pls. make sure that you are in same type of network with your phone setting **(DHCP is defaulted type for the device)**.
Can not dial with regular phone	PSTN or Phone plug not well connection or plug wrong location.	Pls. make sure your line and phone are in good condition. Pls. check the plug in correct location.